AMENDMENTS TO THE CLAIMS

This listing of claims will replace all prior versions and listings of claims in the application:

LISTING OF CLAIMS:

1. (Original) A speech signal decoding method for decoding information concerning at least a sound source signal, gain and linear prediction coefficients from a received signal, generating an excitation signal and linear prediction coefficients from decoded information, and driving a filter, which is constituted by the linear prediction coefficients, by the excitation signal to thereby decode a speech signal, comprising:

a first step of smoothing the gain using a past value of the gain;

a second step of limiting the value of the smoothed gain based upon an amount of fluctuation caculated from the gain and the smoothed gain; and

a third step of decoding the speech signal using the gain that has been smoothed and limited.

2. (Original) A speech signal decoding method for decoding information concerning an excitation signal and linear prediction coefficients from a received signal, generating an excitation signal and linear prediction coefficients from the decoded information, and driving a filter, which is constituted by the linear prediction coefficients, by the excitation signal to thereby decode a speech signal, comprising:

a first step of deriving a norm of the excitation signal at regular intervals;

a second step of smoothing the norm using a past value of the norm;

a third step of limiting the value of the smoothed norm based upon an amount of fluctuation calculated from the norm and the smoothed norm;

a fourth step of changing the amplitude of the excitation signal in said intervals using said norm and the norm that has been smoothed and limited; and

a fifth step of driving the filter by the excitation signal the amplitude of which has been changed.

3. (Currently Amended) A speech signal decoding method for decoding information concerning an excitation signal and linear prediction coefficients from a received signal, generating the excitation signal and the linear prediction coefficients from the decoded information, and driving a filter, which is constituted by the linear prediction coefficients, by the excitation signal to thereby decode a speech signal, comprising:

a first step of identifying a <u>speech</u> ~~voiced~~ segment and a noise segment with regard to the received signal using the decoded information;

a second step of deriving a norm of the excitation signal at regular intervals in the noise segment;

a third step of smoothing the norm using a past value of the norm;

a fourth step of limiting the value of the smoothed norm based upon an amount of fluctuation derived from the norm and the smoothed norm;

a fifth step of changing the amplitude of the excitation signal in said intervals using the norm and the norm that has been smoothed and limited; and

a sixth step of driving the filter by the excitation signal the amplitude of which has been changed.

4. (Original) The method according to claim 1, wherein the amount of fluctuation is represented by dividing an absolute value of a difference between the gain and the smoothed gain by the gain, and the value of the smoothed gain i s limited i n such a manner that the amount of fluctuation will not exceed a predetermined threshold value.

5. (Original) The method according to claim 2, wherein the amount of fluctuation is represented by dividing an absolute value of a difference between the norm and the smoothed norm by the norm, and the value of the smoothed norm is limited in such a manner that the amount of fluctuation will not exceed a predetermined threshold value.

6. (Original) The method according to claim 3, wherein the amount of fluctuation is represented by dividing an absolute value of a difference between the norm and the smoothed

norm by the norm, and the value of the smoothed norm i s limited in such a manner that the amount of fluctuation will not exceed a predetermined threshold value.

7. (Original) The method according to claim 2, wherein the excitation signal in said intervals is divided by the norm in said intervals and the quotient is multiplied by the smoothed norm in said intervals to thereby change the amplitude of the excitation signal.

8. (Original) The method according to claim 3, wherein the excitation signal in said intervals is divided by the norm in said intervals and the quotient is multiplied by the smoothed norm in said intervals to thereby change the amplitude of the excitation signal.

9. (Original) The method according to claim 1, wherein switching between use of the gain and use of the smoothed gain is performed in accordance with an entered switching control signal when the speech signal is decoded.

10. (Original) The method according claim 2, wherein switching between use of the excitation signal and use of the excitation signal the amplitude of which has been changed is performed in accordance with an entered switching control signal when the speech signal is decoded.

11. (Original) The method according claim 3, wherein switching between use of the excitation signal and use of the excitation signal the amplitude of which has been changed is performed in accordance with an entered switching control signal when the speech signal is decoded.

12. (Original) A speech signal encoding and decocting method comerising the steps of:

encoding an input speech signal by expressing the input speech signal by an excitation signal and linear prediction coefficients; and

performing decoding by the speech signal decoding method set forth in claim 1.

13. (Original) A speech signal encoding and decoding method comprising the steps of:

encoding an input speech signal by expressing the input speech signal by an excitation signal and linear prediction coefficients; and

performing decoding by the speech signal decoding method set forth in claim 2.

14. (Original) A speech signal encoding and decoding method comprising the steps of:

encoding an input speech signal by expressing the input speech signal by an excitation signal and linear prediction coefficients; and

performing decoding by the speech signal decoding method set forth in claim 3.

15. (Original) A speech signal decoding apparatus for decoding information concerning at least a sound source signal, gain and linear prediction coefficients from a received signal, generating an excitation signal and linear prediction coefficients from the decoded information, and driving a filter, which is constituted by the linear prediction coefficients, by the excitation signal to thereby decode a speech signal, comprising:

a smoothing circuit smoothing the gain using a past value of the gain; and

a smoothing-quantity limiting circuit limiting the value of the smoothed gain based upon an amount of fluctuation calculated from the gain and the smoothed gain.

16. (Original) A speech signal decoding apparatus for decoding information concerning an excitation signal and linear prediction coefficients from a received signal, generating the excitation signal and linear prediction coefficients from the decoded information, and driving a filter, which is constituted by the linear prediction coefficients, by the excitation signal to thereby decode a speech signal, comprising:

an excitation-signal normalizing circuit deriving a norm of the excitation signal at regular intervals and dividing the excitation signal by the norm;

a smoothing circuit smoothing the norm using a past value of the norm;

a smoothing-quantity limiting circuit limiting the value of the smoothed norm based upon an amount of fluctuation calculated from the norm and the smoothed norm; and

an excitation-signal reconstruction circuit multiplying the smoothed and limited norm by the excitation signal to thereby change the amplitude of the excitation signal in said intervals.

17. (Currently Amended) A speech signal decoding apparatus for decoding information concerning an excitation signal and linear prediction coefficients from a received signal, generating the excitation signal and linear prediction coefficients from the decoded information, and driving a filter, which is constituted by the linear prediction coefficients, by the excitation signal to thereby decode a speech signal, comprising:

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a voiced/unvoiced identification circuit identifying a ~~voiced~~ speech segment and a noise segment with regard to the received signal using the decoded information;

an excitation-signal normalizing circuit deriving a norm of the excitation signal at regular intervals and dividing the excitation signal by the norm;

a smoothing circuit smoothing the norm using a past value of the norm;

a smoothing-quantity limiting circuit limiting the value of the smoothed norm based upon an amount of fluctuation calculated from the norm and the smoothed norm; and

an excitation-signal reconstruction circuit multiplying the smoothed and limited norm by the excitation signal to thereby change the amplitude of the excitation signal in said intervals.

18. (Original) The apparatus according to claim 15, wherein the amount of fluctuation is represented by dividing an absolute value of a difference between the gain and the smoothed

gain by the gain, and the value of the smoothed gain is limited in such a manner that the amount of fluctuation will not exceed a predetermined threshold value.

19. (Original) The apparatus according to claim 16, wherein the amount of fluctuation is represented by dividing the absolute value of the difference between the norm and the smoothed norm by the norm, and the value of the smoothed norm is limited in such a manner that the amount of fluctuation will not exceed a predetermined threshold value.

20. (Original) The apparatus according to claim 17, wherein the amount of fluctuation is represented by dividing the absolute value of the difference between the norm and the smoothed norm by the norm, and the value of the smoothed norm is limited in such a manner that the amount of fluctuation will not exceed a predetermined threshold value.

21. (Original) The apparatus according to claim 15, wherein the apparatus comprises a switching circuit in which switching between use of the gain and use of the smoothed gain is performed in accordance with an entered switching control signal when the speech signal is decoded.

22. (Original) The apparatus according to claim 16, wherein the apparatus comprises a switching circuit in which switching between use of the excitation signal and use of the

excitation signal the amplitude of which has been changed is performed in accordance with an entered switching control signal when the speech signal is decoded.

23. (Original) The apparatus according to claim 17, wherein the apparatus comprises a switching circuit in which switching between use of the excitation signal and use of the excitation signal the amplitude of which has been changed is performed in accordance with an entered switching control signal when the speech signal is decoded.

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24. (Original) A speech signal encoding and decoding apparatus comprising:

a speech signal encoder encoding an input speech signal by expressing the input speech signal by an excitation signal and linear prediction coefficients; and

the speech signal decoding apparatus set forth in claim 15.

25. (Original) A speech signal encoding and decoding apparatus comprising:

a speech signal encoder encoding an input speech signal by expressing the input speech signal by an excitation signal and linear prediction coefficients; and

the speech signal decoding apparatus set forth in claim 16.

26. (Original) A speech signal encoding and decoding apparatus comprising:

a speech signal encoder encoding an input speech signal by expressing the input speech signal by an excitation signal and linear prediction coefficients; and

the speech signal decoding apparatus set forth in claim 17.

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27. (Currenrtly Amended) A program product for causing a computer to execute processing (a) and (b) below, wherein the computer constitutes a speech signal decoding apparatus for decoding information concerning at least a sound source signal, gain and linear prediction coefficients from a received signal, generating an excitation signal and linear prediction coefficients from the decoded information, and driving a filter, which is constituted by the linear prediction coefficients, by the excitation signal to thereby decode a speech signal:

(a) a processing ~~ing of~~ for performing smoothing using a past value of a gain and calculating an amount of fluctuation between the gain and a smoothed gain; and

(b) a processing ~~ing of~~ for -limiting the value of the smoothed gain in conformity with the value of the amount of fluctuation and decoding the speech signal using the smoothed, limited gain.

28. (Currently Amended) A program product for causing a computer to execute processing (a) to (c) below, wherein the computer constitutes a speech signal decoding apparatus for decoding information concerning an excitation signal and linear prediction coefficients from

a received signal, generating an excitation signal and linear prediction coefficients from the decoded information, and driving a filter, which is constituted by the linear prediction coefficients, by the excitation signal to thereby decode a speech signal:

(a) ~~processing of~~ calculating a norm of an excitation signal at regular intervals and smoothing the norm using a past value of the norm;

(b) ~~processing of~~ limiting the value of the smoothed norm in conformity with the value of an amount of fluctuation calculated from the norm and the smoothed norm; and

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(c) ~~processing of~~ changing the amplitude of the excitation signal in said intervals using the norm and the norm that has been smoothed and limited, and driving the filter by the excitation signal the amplitude of which has been changed.

29. (Currently Amended) A program product for causing a computer to execute processing (a) to (d) below, wherein the computer constitutes a speech signal decoding apparatus for decoding information concerning an excitation signal and linear prediction coefficients from a received signal, generating an excitation signal and linear prediction coefficients from the decoded information, and driving a filter, which is constituted by the linear prediction coefficients, by the excitation signal to thereby decode a speech signal:

(a) ~~processing of~~ identifying a <u>speech</u> ~~voiced~~ segment and a noise segment with regard to a received signal using decoded information;

(b) ~~processing of~~ calculating a norm of an excitation signal at regular intervals in the noise segment and smoothing the norm using a past value of the norm;

(c) ~~processing of~~ limiting the value of the smoothed norm in conformity with an amount of fluctuation calculated from the norm and the smoothed norm; and

(d) ~~processing of~~ changi ng the amplitude of the excitation signal in said intervals using the norm and the norm that has been smoothed and limited, and driving the filter by the excitation signal the amplitude of which has been changed.

30. (Currently Amended) The program product according to claim 27, wherein said program product comprises a program for a processing of representing the amount of fluctuation by dividing an absolute value of a difference between the gain and the smoothed gain by the gain, and limiting the value of the smoothed gain in such a manner that the amount of fluctuation will not exceed a predetermined threshold value.

31. (Currently Amended) The program product according to claim 28, wherein said program product comprises a program ~~for~~ to execute a processing of representing the amount of fluctuation by dividing an absolute value of a difference between the norm and the smoothed norm by the norm, and limiting the value of the smoothed norm in such a manner that the amount of fluctuation will not exceed a predetermined threshold value.

32. (Currently Amended) The program product according to claim 29, wherein said program product comprises a program ~~for processing of~~ representing the amount of fluctuation by dividing an absolute value of a difference between the norm and the smoothed norm by the norm, and limiting the value of the smoothed norm in such a manner that the amount of fluctuation will not exceed a predetermined threshold value.

33. (Currently Amended) The program product according to claim 28, wherein said program product comprises a program ~~for processing of~~ dividing the excitation signal in said intervals by the norm in said intervals and multiplying the quotient by the smoothed norm in said intervals to thereby change the amplitude of the excitation signal.

34. (Currently Amended) The program product according to claim 29, wherein said program product comprises a program ~~for processing of~~ dividing the excitation signal in said intervals by the norm in said intervals and multiplying the quotient by the smoothed norm in said intervals to thereby change the amplitude of the excitation signal.

35. (Currently Amended) The program product according to claim 27, wherein said program product comprises a program ~~for processing of~~ switching between use of the gain and use the smoothed gain i n accordance with an entered switching control signal when the speech signal is decoded.

36. (Currently Amended) The program product according to claim 28, wherein said program product comprises a program ~~for processing of~~ switching between use of the excitation signal and use of the excitation signal the amplitude of which has been changed in accordance with an entered switching control signal when the speech signal is decoded.

37. (Currently Amended) The program product according to claim 29, wherein said program product comprises a program ~~for processing of~~ switching between use of the excitation signal and use of the excitation signal the amplitude of which has been changed in accordance with an entered switching control signal when the speech signal is decoded.

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38. (Currently Amended) A program product comprising a program for causing said computer to execute ~~processing of performing~~ decoding by the speech signal decoding method set forth in claim 1, when an input speech signal has been encoded by expressing the input speech signal by an excitation signal and linear prediction coefficients.

39. (Currently Amended) A program product comprising a program for causing said computer to execute ~~processing of performing~~ decoding by the speech signal decoding method set forth in claim 2, when an input speech signal has been encoded by expressing the input speech signal by an excitation signal and linear prediction coefficients.

40. (Currently Amended) A program product comprising a program for causing said computer to execute ~~processing of performing~~ decoding by the speech signal decoding method set forth in claim 3, when an input speech signal has been encoded by expressing the input speech signal by an excitation signal and linear prediction coefficients.

41. (Original) A speech signal decoding apparatus comprising:

(a) a code input circuit splitting code of a bit sequence of an encoded input signal that enters from an input terminal, converting the code to indices that correspond to a plurality of decode parameters, outputting an index corresponding to a line spectrum pair, termed hereinafter "LSP", which represents the frequency characteristic of the input signal, to an LSP decoding circuit, outputting an index corresponding to a delay that represents a pitch period of the input signal to a pitch signal decoding circuit, outputting an index corresponding to a sound source vector comprising a random number or a pulse train to a sound source signal decoding circuit, outputting an index corresponding to a first gain to a first gain decoding circuit, and outputting an index corresponding to a second gain to a second gain decoding circuit;

(b) an LSP decoding circuit, to which the index output from said code input circuit is input, and which reads the LSP corresponding to the input index out of a table which stores LSPs corresponding to indices, obtains an LSP in a subframe of the present frame and outputs the LSP;

(c) a linear prediction coefficient conversion circuit, to which the LSP output from said LSP decoding circuit is input, and which converts the LSP to linear prediction coefficients and outputs the coefficients to a synthesis filter;

(d) a sound source signal decoding circuit, to which the index output from said code input circuit is input, and which reads a sound source vector corresponding to the index out of a table storing sound source vectors corresponding to indices, and outputs the sound source vector to a second gain decoding circuit;

(e) a second gain decoding circuit, to which the index output from said code input circuit is input, and which reads a second gain corresponding to the input index out of a table storing second gains corresponding to indices, and outputs the second gain to a smoothing circuit;

(f) a second gain circuit, to which a first sound source vector output from said sound source signal decoding circuit and the second gain are input, and which multiplies the first sound source vector by the second gain to generate a second sound source vector and outputs the generated second sound source vector to an adder;

(g) a memory circuit holding an excitation vector input thereto from said adder and outputting a held excitation vector, which was input thereto in the past, to a pitch signal decoding circuit;

(h) a pitch signal decoding circuit, to which the past excitation vector held by said memory circuit and the index output from said code input circuit are input, with said index specifying a delay, and which cuts out vectors of samples corresponding to a vector length from

a point previous to the starting point of the present frame by an amount corresponding to the delay to thereby generate a first pitch vector, and outputs the first pitch vector to a first gain circuit;

(i) a first gain decoding circuit, to which the index output from said code input circuit is input, and which reads a first gain corresponding to the input index out of a table storing first gains corresponding to indices, and outputs the first gain to a first gain circuit;

(j) a first gain circuit, to which the first pitch vector output from said pitch signal decoding circuit and the first gain output from said first gain decoding circuit are input, and which multiplies the input first pitch vector by the first gain to generate a second pitch vector, and outputs the generated second pitch vector to said adder;

(k) an adder, to which the second pitch vector output from said first gain circuit and the second sound source vector output from said second gain circuit are input, and which calculates the sum of these inputs, and outputs the sum to a synthesis filter as an excitation vector;

(l) a smoothing coefficient calculation circuit, to which LSP output from said LSP decoding circuit is input, and which calculates average LSP in the present frame, finds the amount of fluctuation of the LSP with respect to each subframe, finds a smoothing coefficient in the subframe, and outputs the smoothing coefficient to a smoothing circuit;

(m) a smoothing circuit, to which the smoothing coefficient output from said smoothing coefficient calculation circuit and the second gain output from said second gain decoding circuit

are input, and which finds an average gain from the second gain in the subframe, and outputs the second gain;

(n) a synthesis filter, to which the excitation vector output from said adder and the linear prediction coefficients output from said linear prediction coefficient conversion circuit are input, and which drives a synthesis filter, for that the linear prediction coefficients have been set, by the excitation vector to thereby calculate a reconstructed vector, and outputs the reconstructed vector from an output terminal; and

(o) a smoothing-quantity limiting circuit, to which the second gain output from said second gain decoding circuit and the smoothed second gain output from said smoothing circuit are input, and which finds the amount of fluctuation between the smoothed second gain output from said smoothing circuit and the second gain output from said second gain decoding circuit,

outputs the smoothed second gain to said second gain circuit as is when the amount of fluctuation is less than a predetermined threshold value, replaces the smoothed second gain with a smoothed second gain limited in terms of values it is capable of taking on when the amount of fluctuation is equal to or greater than the threshold value, and outputs this smoothed second gain to said second gain circuit.

42. (Original) The apparatus according to claim 41, further comprising:

(p) an excitation-signal normalizing circuit, to which an excitation vector in a subframe output from said adder is input, and which calculates gain and a shape vector from the excitation

vector every subframe or every sub-subframe obtained by subdividing a subframe, outputs the gain to said smoothing circuit, and outputs the shape vector to an excitation-signal reconstruction circuit; and

(q) an excitation-signal reconstruction circuit, to which the gain output from said smoothing-quantity limiting circuit and the shape vector output from said excitation-signal normalizing circuit are input, and which calculates a smoothed excitation vector, and outputs this excitation vector to said memory circuit and to said synthesis filter;

(r) wherein said smoothing circuit has the output of said excitation-signal normalizing circuit input thereto instead of the output of said second gain decoding circuit and has the output of said smoothing coefficient calculation circuit input thereto;

(s) said smoothing-quantity limiting circuit has the smoothed gain output from said smoothing circuit applied to one input terminal thereof and has the gain output from said

excitation-signal normalizing circuit, rather than the output of said second gain decoding circuit, applied to the other input terminal thereof, finds the amount of fluctuation between the smoothed gain output from said smoothing circuit and the gain output from said excitation-signal normalizing circuit, supplies the smoothed gain as is to said excitation-signal reconstruction circuit when the amount of fluctuation is less than a predetermined threshold value, replaces the smoothed gain with a smoothed gain limited in terms of values it is capable of taking on when the amount of fluctuation is equal to or greater than the threshold value, and supplies this smoothed gain to the excitation-signal reconstruction circuit; and

(t) the output of said second gain decoding circuit is input to said second gain circuit as second gain.

43. (Currently Amended) The apparatus according to claim 42, further comprising:

a power calculation circuit, to which the reconstructed vector output from said synthesis filter is input, and which calculates the sum of the squares of the reconstructed vector and outputting the power to a voiced/unvoiced identification circuit;

a speech mode decision circuit, to which a past excitation vector held by said memory circuit and an index specifying a delay output from said code input circuit are input, and which calculates a pitch prediction gain in a subframe from the past excitation vector and the delay, determines a predetermined threshold value with respect to the pitch prediction gain or with respect to an in-frame average value of the Pitch prediction gain in a certain frame, and sets a speech mode;

a voiced/unvoiced identification circuit, to which an LSP output from said LSP decoding circuit, the speech mode output f rom said speech mode decision circuit and the power output from said power calculation circuit are input, and which finds the amount of fluctuation of a spectrum parameter, identifying a speech~~voice~~ segment and noise ~~an unvoiced~~ segment based upon the amount of fluctuation, and outputs amount-of-fluctuation information and an identification flag;

a noise classification circuit, to which the amount-of-fluctuation information and identification flag output from said voiced/unvoiced identification are input, and which classifies noise and outputting a classification flag; and

a first changeover circuit, to which the gain output from said excitation-signal normalizing circuit, the identification flag output from said voiced/unvoiced identification circuit and the classification flagoutput from the noise classification circuit are input, and which changes over a sw i tch i n accordance with a value of the identification flag and a value of the classification flag to thereby switchingly output the gain to any one of a plurality of filters having different filter characteristics from one another;

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wherein the filter selected from among said plurality of filters has the gain output from said first changeover circuit applied thereto, smoothes the gain using a linear filter or non-linear filter and outputs the smoothed gain to said smoothing-quantity limiting circuit as a first smoothed gain; and

said smoothing-quantity limiting circuit has the first smoothed gain output from the selected filter applied to one input terminal thereof, has the output of said excitation-signal normalizing circuit applied to the other input terminal thereof, finds the amount of fluctuation between the gain output from said excitation-signal normalizing circuit and the first smoothed gain output from said selected filter, uses the first smoothed gain as is when the amount of fluctuation is less than a predetermined threshold value, replaces the first smoothed gain with a smoothed gain limited in terms of values it is capable of taking on when the amount of

fluctuation is equal to or greater than the threshold value, and supplies this smoothed gain to said excitation-signal reconstruction circuit.

44. (Original) The apparatus according to claim 41, further comprising a changeover circuit switching between a mode of using of the gain and a mode of using the smoothed gain as the input to said second gain circuit in accordance with a switching control signal, which has entered from an input terminal, when the speech signal is decoded.

45. (Original) The apparatus according to claim 42, further comprising a changeover circuit to which the excitation vector output from said adder is input, and which outputs the excitation vector to said synthesis filter or to said excitation-signal normalizing circuit in accordance with a changeover control signal, that has entered from an input terminal.

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46. (Original) The apparatus according to claim 43, further comprising a changeover circuit to which the excitation vector output from aid adder is input, and which outputs the excitation vector to said synthesis filter or to said excitation-signal normalizing circuit in accordance with a changeover control signal, that has entered from an input terminal.

47. (New) A speech signal decoding method for decoding information concerning at least a sound source signal, gain and linear prediction coefficients from a received signal,

generating an excitation signal and linear prediction coefficients from decoded information, and driving a filter, which is constituted by the linear prediction coefficients, by the excitation signal to thereby decode a speech signal, comprising:

a first step of smoothing the gain using a past value of the gain;

a second step of limiting the value of the smoothed gain based upon the past smoothed gain; and

a third step of decoding the speech signal using the gain that has been smoothed and limited.

48. (New) A speech signal decoding method for decoding Information concerning an excitation signal and linear prediction coefficients from a received signal, generating an excitation signal and linear prediction coefficients from the decoded information, and driving a filter, which is constituted by the linear prediction coefficients, by the excitation signal to thereby decode a speech signal, comprising:

a first stop of deriving a norm of the excitation signal at regular intervals;

a second step of smoothing the norm using a past value of the norm;

a third step of limiting the value of the smoothed norm based upon the past smoothed norm;

a fourth step of changing the amplitude of the excitation signal in said intervals using said norm and the norm that has been smoothed and limited; and

a fifth step of driving the filter by the excitation signal the amplitude of which has been

changed.

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